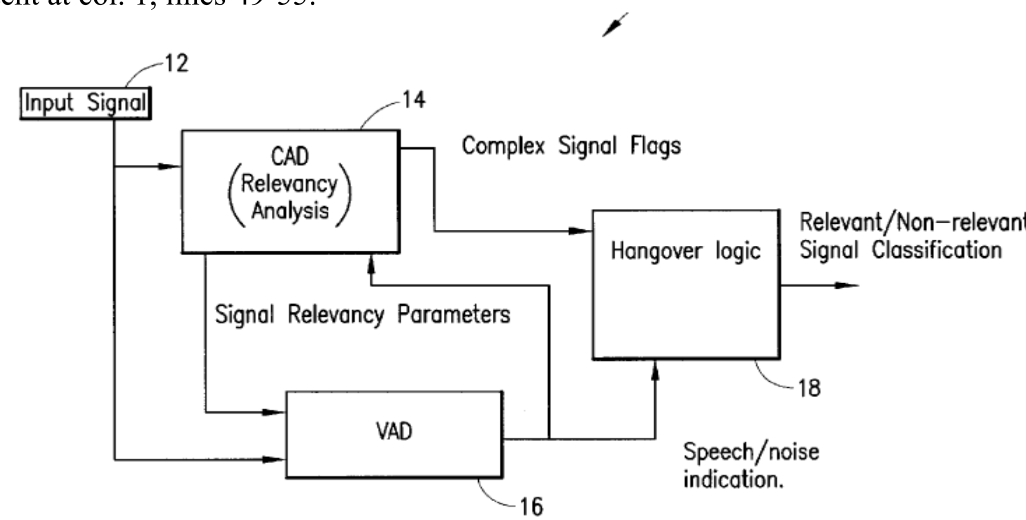
APPENDIX 1-A
U.S. Patent 6,633,841 IS INVALID UNDER 35 U.S.C. § 102 IN LIGHT OF U.S. Patent 6,424,938

Claims	Invalidity based on US 6,424,938
1[a] An extended signal codec that performs signal coding of a speech signal, the extended signal codec comprising:	U.S. Patent No. 6,424,938 ("the '938 Patent") discloses an extended signal codec that performs signal coding of a speech signal. For example, and without limitation: "FIG. 1 diagrammatically illustrates pertinent portions of exemplary embodiments of a speech encoding apparatus according to the invention. The speech encoding apparatus can be provided, for example, in a radio transceiver that communicates audio information via a radio communication channel. One example of such a radio transceiver is a mobile radiotelephone such as a cellular telephone." '938 Patent at col. 1, lines 49-55.  '938 Patent, FIG. 1.
[1b] a background noise speech signal coding module;	The '938 Patent discloses a codec that includes a background noise speech signal coding module. For example, and without limitation: "In compressing the incoming audio signal, speech encoders conventionally use advanced lossy compression techniques. The compressed (or coded) signal information is transmitted to the decoder via a communication channel such as a radio link. The decoder then attempts to reproduce the input audio signal from the compressed signal information. If certain characteristics of the incoming audio signal are known, then the bit rate in the communication channel can be maintained as low as possible. If the audio signal contains relevant information for the listener, then this information should be retained. However,

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	if the audio signal contains only irrelevant information (for example background noise), then bandwidth can be saved by only transmitting a limited amount of information about the signal. For many signals which contain only irrelevant information, a very low bit rate can often provide high quality compression. In extreme cases, the incoming signal may be synthesized in the decoder without any information updates via the communication channel until the input audio signal is again determined to include relevant information. Typical signals which can be conventionally reproduced quite accurately with very low bit rates include stationary noise, car noise and also, to some extent, babble noise. More complex non-speech signals like music, or speech and music combined, require higher bit rates to be reproduced accurately by the decoder. For many common types of background noise a much lower bit rate than is needed for speech provides a good enough model of the signal. Existing mobile systems make use of this fact by downwardly adjusting the transmitted bit rate during background noise. For example, in conventional systems using continuous transmission techniques, a variable rate (VR) speech coder may use its lowest bit rate. In conventional Discontinuous Transmission (DTX) schemes, the transmitter stops sending coded speech frames when the speaker is inactive. At regular or irregular intervals (for example, every 100 to 500 ms), the transmitter sends speech parameters suitable for conventional generation of comfort noise in the decoder. These parameters for comfort noise generation (CNG) are conventionally coded into what are sometimes called Silence Descriptor (SID) frames. At the receiver, the decoder uses the comfort noise parameters received in the SID frames to synthesize artificial noise by means of a conventional comfort noise injection (CNI) algorithm." '938 Patent at col. 1, line 32 – col. 2, line 10. "Conventional classification techniques for determining whether or not an input audio signal contains relevant information are primarily based on a relatively simple stationarity analysis of the input audio signal. If the input signal is determined to be stationary, then it is assumed to be a noise-like signal. However, this conventional stationarity analysis alone can cause complex signals that are fairly stationary but actually contain perceptually relevant information to be misclassified as noise. Such a misclassification disadvantageously results in the problems described above.

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	It is therefore desirable to provide a classification technique that reliably detects the presence of perceptually relevant information in complex signals of the type described above. According to the present invention, complex signal activity detection is provided for reliably detecting complex non-speech signals that include relevant information that is perceptually important to the listener. Examples of complex non-speech signals that can be reliably detected include music, music on-hold, speech and music combined, music in the background, and other tonal or harmonic sounds.” ‘938 Patent at col. 2, line 60 – col. 3, line 13. “In FIG. 1, the input audio signal is input to a complex signal activity detector (CAD) and also to a voice activity detector (VAD). The complex signal activity detector CAD is responsive to the audio input signal to perform a relevancy analysis that determines whether the input signal includes information that is perceptually relevant to the listener, and provide a set of signal relevancy parameters to the VAD. The VAD uses these signal relevancy parameters in conjunction with the received audio input signal in order to determine whether the input audio signal is speech or noise. The VAD operates as a speech/noise classifier; and provides as an output a speech/noise indication. The CAD receives the speech/noise indication as an input. The CAD is responsive to the speech/noise indication and the input audio signal to produce a set of complex signal flags which are output to a hangover logic section which also receives as an input the speech/noise indication provided by the VAD.” ‘938 Patent at col. 3, line 56 – col. 4, line 5.

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	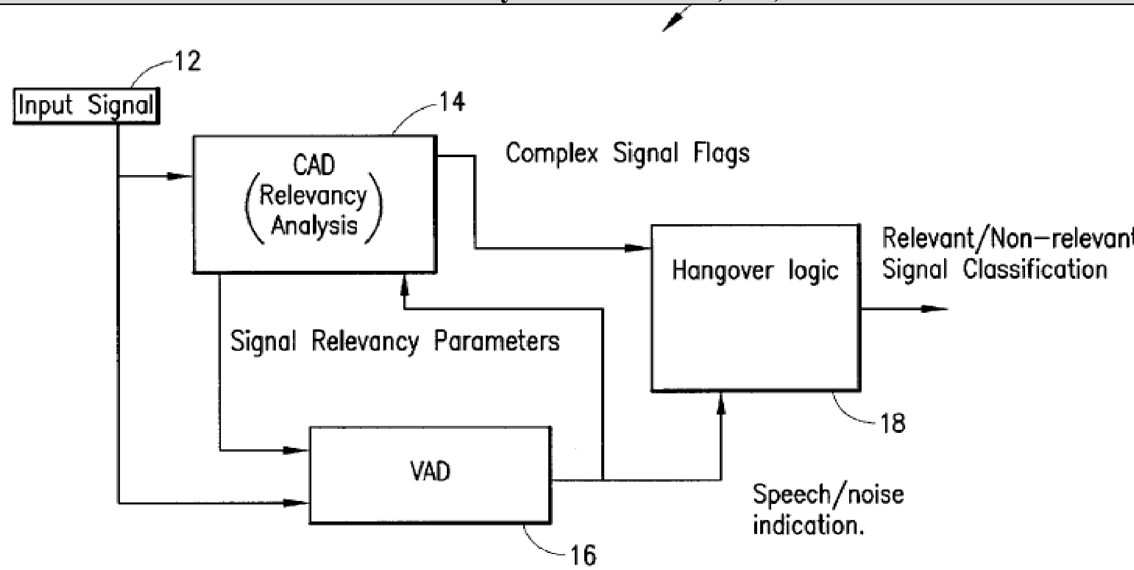 '938 Patent, FIG. 1. "From the foregoing description, it will be evident to workers in the art that the embodiments of FIGS. 1-13 can be readily implemented by suitable modifications in software, hardware, or both, in a conventional speech encoding apparatus." '938 Patent at col. 10, lines 31-35.
[1c] a music speech signal coding module;	The '938 Patent discloses a music speech signal coding module. For example, and without limitation: "In compressing the incoming audio signal, speech encoders conventionally use advanced lossy compression techniques. The compressed (or coded) signal information is transmitted to the decoder via a communication channel such as a radio link. The decoder then attempts to reproduce the input audio signal from the compressed signal information. If certain characteristics of the incoming audio signal are known, then the bit rate in the communication channel can be maintained as low as possible. If the audio signal contains relevant information for the listener, then this information should be retained. However,

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	if the audio signal contains only irrelevant information (for example background noise), then bandwidth can be saved by only transmitting a limited amount of information about the signal. For many signals which contain only irrelevant information, a very low bit rate can often provide high quality compression. In extreme cases, the incoming signal may be synthesized in the decoder without any information updates via the communication channel until the input audio signal is again determined to include relevant information. Typical signals which can be conventionally reproduced quite accurately with very low bit rates include stationary noise, car noise and also, to some extent, babble noise. More complex non-speech signals like music, or speech and music combined, require higher bit rates to be reproduced accurately by the decoder." '938 Patent at col. 1, lines 32-57. "If a complex signal like music is compressed using a compression model that is too simple, and a corresponding bit rate that is too low, the reproduced signal at the decoder will differ dramatically from the result that would be obtained using a better (higher quality) compression technique. The use of a too simple compression scheme can be caused by misclassifying the complex signal as noise. When such misclassification occurs, not only does the decoder output a poorly reproduced signal, but the misclassification itself disadvantageously results in a switch from a higher quality compression scheme to a lower quality compression scheme. To correct the misclassification, another switch back to the higher quality scheme is needed. If such switching between compression schemes occurs frequently, it is typically very audible and can be irritating to the listener. It can be seen from the foregoing that it is desirable to reduce the misclassification of subjectively relevant signals, while still maintaining a low bit rate (high compression) where appropriate, for example when compressing background noise while the speaker is silent. Very strong compression techniques can be used, provided they are not perceived as irritating. The use of comfort noise parameters as described above with respect to DTX systems is an example of a strong compression technique, as is conventional low rate linear predictive coding (LPC) using random excitation methods. Coding techniques such as these, which utilize strong compression, can typically reproduce accurately only perceptually simple noise types such as stationary car noise, street noise, restaurant noise (babble) and other similar signals.

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	Conventional classification techniques for determining whether or not an input audio signal contains relevant information are primarily based on a relatively simple stationarity analysis of the input audio signal. If the input signal is determined to be stationary, then it is assumed to be a noise-like signal. However, this conventional stationarity analysis alone can cause complex signals that are fairly stationary but actually contain perceptually relevant information to be misclassified as noise. Such a misclassification disadvantageously results in the problems described above. It is therefore desirable to provide a classification technique that reliably detects the presence of perceptually relevant information in complex signals of the type described above. According to the present invention, complex signal activity detection is provided for reliably detecting complex non-speech signals that include relevant information that is perceptually important to the listener. Examples of complex non-speech signals that can be reliably detected include music, music on-hold, speech and music combined, music in the background, and other tonal or harmonic sounds." '938 Patent at col. 2, line 29 – col. 3, line 13. "In FIG. 1, the input audio signal is input to a complex signal activity detector (CAD) and also to a voice activity detector (VAD). The complex signal activity detector CAD is responsive to the audio input signal to perform a relevancy analysis that determines whether the input signal includes information that is perceptually relevant to the listener, and provide a set of signal relevancy parameters to the VAD. The VAD uses these signal relevancy parameters in conjunction with the received audio input signal in order to determine whether the input audio signal is speech or noise. The VAD operates as a speech/noise classifier; and provides as an output a speech/noise indication. The CAD receives the speech/noise indication as an input. The CAD is responsive to the speech/noise indication and the input audio signal to produce a set of complex signal flags which are output to a hangover logic section which also receives as an input the speech/noise indication provided by the VAD." '938 Patent at col. 3, line 56 – col. 4, line 5.

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	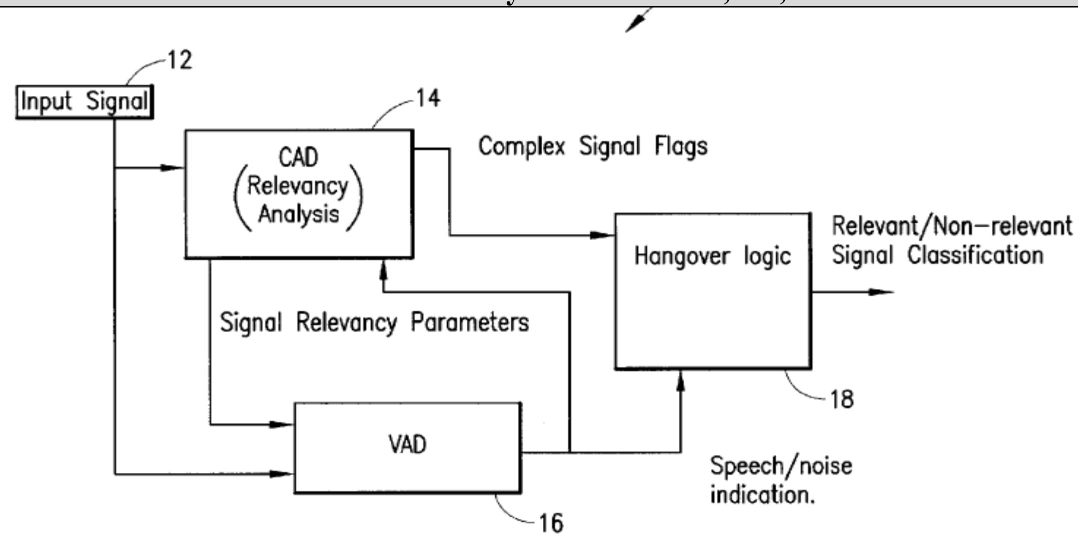 '938 Patent, FIG. 1. "From the foregoing description, it will be evident to workers in the art that the embodiments of FIGS. 1-13 can be readily implemented by suitable modifications in software, hardware, or both, in a conventional speech encoding apparatus." '938 Patent at col. 10, lines 31-35.

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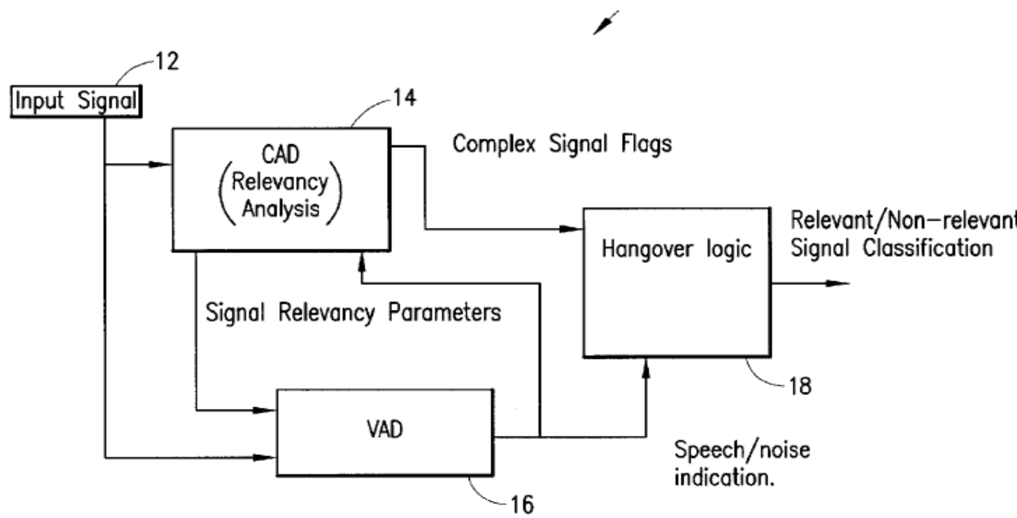
[1d] a voice activity detection module configured to generate a decision signal, wherein the decision signal is of a first type if the voice activity detection module detects no voice activity in the speech signal or of a second type if the voice activity detection module detects voice activity in the speech signal, and wherein the first type is associated with selection of the background noise speech signal coding module and the second type is associated with selection of the music speech signal coding module; and	The '938 Patent discloses a voice activity detection module (VAD 16) configured to generate a decision signal (speech/noise indication), wherein the decision signal is of a first type if the voice activity detection module detects no voice activity in the speech signal or of a second type if the voice activity detection module detects voice activity in the speech signal, and wherein the first type is associated with selection of the background noise speech signal coding module and the second type is associated with selection of the music speech signal coding module. For example, and without limitation:  '938 Patent, FIG. 1. "In FIG. 1, the input audio signal is input to a complex signal activity detector (CAD) and also to a voice activity detector (VAD). The complex signal activity detector CAD is responsive to the audio input signal to perform a relevancy analysis that determines whether the input signal includes information that is perceptually relevant to the listener, and provide a set of signal relevancy parameters to the VAD. The VAD uses these signal relevancy parameters in conjunction with the received audio input signal in order to determine whether the input audio signal is speech or noise. The VAD operates as a speech/noise classifier; and provides as an output a speech/noise indication. The CAD receives the speech/noise indication as an input. The CAD is responsive to the speech/noise indication and the input audio signal to produce a set of complex signal flags which are output to a hangover logic section which also receives as

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	an input the speech/noise indication provided by the VAD. The hangover logic is responsive to the complex signal flags and the speech/noise indication for providing an output which indicates whether or not the input audio signal includes information which is perceptually relevant to a listener who will hear a reproduced audio signal output by a decoding apparatus in a receiver at the other end of the communication channel. The output of the hangover logic can be used appropriately to control, for example, DTX operation (in a DTX system) or the bit rate (in a variable rate VR encoder). If the hangover logic output indicates that input audio signal does not contain relevant information, then comfort noise can be generated (in a DTX system) or the bit rate can be lowered (in a VR encoder). " '938 Patent at col. 3, line 55 – col. 4, line 18. "1. A method of preserving perceptually relevant non-speech information in an audio signal during encoding of the audio signal, comprising: making a first determination of whether the audio signal is considered to comprise speech or noise information; making a second determination of whether the audio signal includes non-speech information that is perceptually relevant to a listener; and selectively overriding said first determination in response to said second determination." '938 Patent claim 1. "13. An apparatus for use in an audio signal encoder to preserve perceptually relative non-speech information contained in an audio signal, comprising: a classifier for receiving the audio signal and making a first determination of whether the audio signal is considered to comprise speech or noise information; a detector for receiving the audio signal and making a second determination of whether the audio signal includes non-speech information that is perceptually relevant to a listener; and logic coupled to said classifier and said detector, said logic having an output for indicating whether the audio signal includes perceptually relevant information, said logic operable to selectively provide at said output information indicative of said first determination, and also responsive to said second determination for selectively overriding at said output said information indicative of said first determination." '938 Patent claim 13.

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	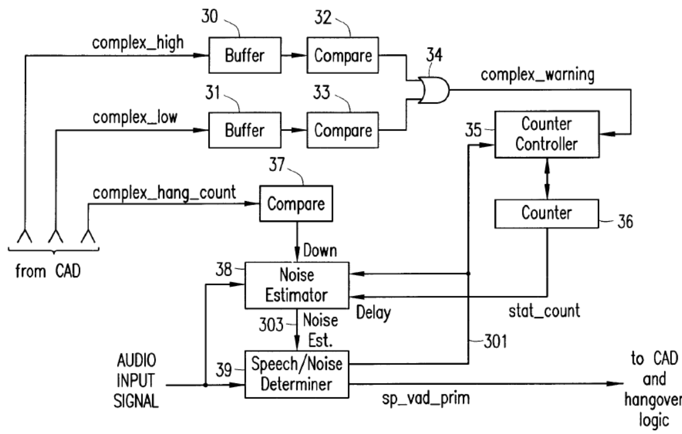 *FIG. 3* '938 Patent Fig. 3. "FIG. 3 illustrates pertinent portions of exemplary embodiments of the VAD of FIG. 1. As described above with respect to FIG. 2, the VAD receives from the CAD signal relevancy parameters complex_high, complex_low and complex_hang_count. Complex_high and complex_low are input to respective buffers 30 and 31, whose outputs are respectively coupled to comparators 32 and 33. The outputs of the comparators 32 and 33 are coupled to respective inputs of an OR gate 34 which outputs a complex_warning signal to a counter controller 35. The counter controller 35 controls a counter 36 in response to the complex_warning signal." '938 Patent at col. 7, lines 25-35.

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[1e] a voice activity detection correction and supervision module configured to receive the decision signal, wherein if the voice activity module generates the decision signal of the first type, the voice activity detection correction and supervision module overrides the decision signal of the first type and generates a new decision signal of the second type if the voice activity detection correction and supervision module detects at least one characteristic of the speech signal indicative of a music signal in the speech signal.	The '938 Patent discloses a voice activity detection correction and supervision module configured to receive the decision signal, wherein if the voice activity module generates the decision signal of the first type, the voice activity detection correction and supervision module overrides the decision signal of the first type and generates a new decision signal of the second type if the voice activity detection correction and supervision module detects at least one characteristic of the speech signal indicative of a music signal in the speech signal. For example, and without limitation: "1. A method of preserving perceptually relevant non-speech information in an audio signal during encoding of the audio signal, comprising: making a first determination of whether the audio signal is considered to comprise speech or noise information; making a second determination of whether the audio signal includes non-speech information that is perceptually relevant to a listener; and selectively overriding said first determination in response to said second determination." '938 Patent claim 1. "13. An apparatus for use in an audio signal encoder to preserve perceptually relative non-speech information contained in an audio signal, comprising: a classifier for receiving the audio signal and making a first determination of whether the audio signal is considered to comprise speech or noise information; a detector for receiving the audio signal and making a second determination of whether the audio signal includes non-speech information that is perceptually relevant to a listener; and logic coupled to said classifier and said detector, said logic having an output for indicating whether the audio signal includes perceptually relevant information, said logic operable to selectively provide at said output information indicative of said first determination, and also responsive to said second determination for selectively overriding at said output said information indicative of said first determination." '938 Patent claim 13. "20. The apparatus of claim 13, wherein said logic is operable for overriding information indicative of a noise determination in response to said second determination indicating perceptually relevant non-speech information." '938 Patent claim 20.

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	"The complex signal activity detector CAD is responsive to the audio input signal to perform a relevancy analysis that determines whether the input signal includes information that is perceptually relevant to the listener, and provide a set of signal relevancy parameters to the VAD. The VAD uses these signal relevancy parameters in conjunction with the received audio input signal in order to determine whether the input audio signal is speech or noise. The VAD operates as a speech/noise classifier; and provides as an output a speech/noise indication. The CAD receives the speech/noise indication as an input. The CAD is responsive to the speech/noise indication and the input audio signal to produce a set of complex signal flags which are output to a hangover logic section which also receives as an input the speech/noise indication provided by the VAD. The hangover logic is responsive to the complex signal flags and the speech/noise indication for providing an output which indicates whether or not the input audio signal includes information which is perceptually relevant to a listener who will hear a reproduced audio signal output by a decoding apparatus in a receiver at the other end of the communication channel. The output of the hangover logic can be used appropriately to control, for example, DTX operation (in a DTX system) or the bit rate (in a variable rate VR encoder). If the hangover logic output indicates that input audio signal does not contain relevant information, then comfort noise can be generated (in a DTX system) or the bit rate can be lowered (in a VR encoder)." '938 Patent at col. 3, line 58 – col. 4, line 18. "The hangover logic adjusts the final decision of the signal using previous information on the relevancy of the signal and the previous VAD decisions, if the VAD is considered to be reliable. The output of the hangover logic is a final decision on whether the signal is relevant or non-relevant. In the non-relevant case a low bit rate can be used for encoding. In a DTX system this relevant/non-relevant information is used to decide whether the present frame should be coded in the normal way (relevant) or whether the frame should be coded with comfort noise parameters (non-relevant) instead." '938 Patent at col. 4, line 65 – col. 5, line 7.

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	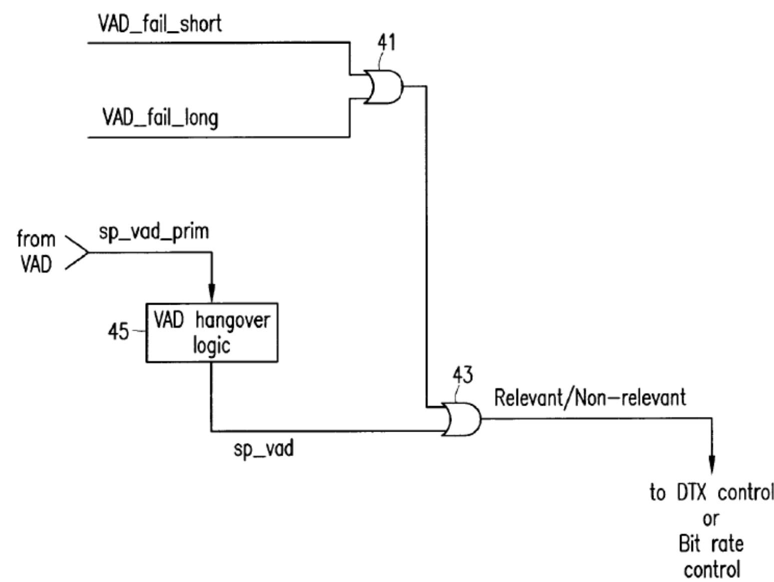 FIG. 4 '938 Patent, Fig. 4. "FIG. 4 illustrates an exemplary embodiment of the hangover logic of FIG. 1. In FIG. 4, the complex signal flags VAD_fail_short and VAD_fail_long are input to an OR gate 41 whose output drives an input of another OR gate 43. The speech/noise indication sp_vad_prim from the VAD is input to conventional VAD hangover logic 45. The output sp_vad of the VAD hangover logic is coupled to a second input of OR gate 43. If either of the complex signal flags VAD_fail_short or VAD_fail_long is active, then the output of OR gate 41 will cause the OR gate 43 to indicate that the input signal is relevant. '938 Patent at col. 8, lines 23-33.

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3. The extended signal codec of claim 1, wherein the at least one characteristic of the speech signal corresponds to a pitch information.	Defendants hereby incorporate by reference the disclosures above for claim 1. The ‘938 patent discloses the extended signal codec of claim 1, wherein the at least one characteristic of the speech signal corresponds to a pitch information. For example, and without limitation: “According to the present invention, complex signal activity detection is provided for reliably detecting complex non-speech signals that include relevant information that is perceptually important to the listener. Examples of complex non-speech signals that can be reliably detected include music, music on-hold, speech and music combined, music in the background, and other tonal or harmonic sounds.” ‘938 Patent at col. 3, lines 7-13. “The input signal (which can be preprocessed) is analyzed in the CAD by extracting information each frame about the correlation of the signal in a specific frequency band. This can be accomplished by first filtering the signal with a suitable filter, e.g., a bandpass filter or a high pass filter. This filter weighs the frequency bands which contain most of the energy of interest in the analysis. Typically, the low frequency region should be filtered out in order to de-emphasize the strong low frequency contents of, e.g., car noise. The filtered signal can then be passed to an open-loop long term prediction (LTP) correlation analysis. The LTP analysis provides as a result a vector of correlation values or normalized gain values; one value per correlation shift. The shift range may be, for example, [20, 147] as in conventional LTP analysis. An alternative, low complexity, method to achieve the desired relevancy detection is to use the unfiltered signal in the correlation calculation and modify the correlation values by an algorithmically similar "filtering" process, as described in detail below. For each analysis frame, the normalized correlation value (gain value) having the largest magnitude is selected and buffered. The shift (corresponding to the LTP lag of the selected correlation value) is not used. The values are further analyzed to provide a vector of Signal Relevancy Parameters which is sent to the VAD for use by the background noise estimation process. The buffered correlation values are also processed and used to make a definitive decision as to whether the signal is relevant (i.e., has perceptual importance) and whether the VAD decision is reliable. A set of flags, VAD_fail_long and VAD_fail_short, are produced to indicate when it is likely that the VAD will make a severe misclassification, that is, a noise classification when perceptually relevant information is in fact present.”

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	‘938 Patent at col. 4, lines 19-52. “2. The method of claim 1, wherein said step of making said second determination includes the additional steps of: determining, from the audio signal, correlation values using an open-loop long term prediction correlation analysis; and comparing a predetermined value to the correlation values associated with respective frames into which the audio signal is divided.” ‘938 Patent, claim 2. It is well known to those of ordinary skill in the art of speech coding that open-loop long-term prediction (LTP) correlation analysis necessarily involves the determination of pitch information in speech signals. Thus, the determination of pitch information is inherent to the open-loop long-term prediction correlation analysis disclosed in the ‘938 Patent. For example, US Patent 6,199,035, filed May 6, 1998, is directed to pitch-lag estimation in speech coding, and explains:

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	The LTP parameters include the so called pitch-lag parameter which describes the fundamental frequency of the speech signal. The determination of the pitch-lag for a current frame of the residual signal is carried out in two stages. Firstly, an open-loop search is conducted, involving a relatively coarse search of the residual signal, subject to a predefined maximum and minimum delay, for a portion of the signal which best matches the current frame. A closed-loop search is then conducted over the already synthesised signal. The closed-loop search is conducted over a small range of delays in the neighbourhood of the open-loop estimate of pitch-lag. It is important to note that if a mistake is made in the open-loop search, the mistake cannot be corrected in the closed-loop search. In early known codecs, the open-loop LTP analysis determines the pitch-lag for a given frame of the residual signal by determining the autocorrelation function of the frame within the residual speech signal, i.e.: $$\hat{R}(d)=\sum_{n=0}^{N-1} r(n-d)r(n) \quad d=d_L, \ldots, d_H$$ where d is the delay, r(n) is the residual signal, and d_L and d_H are the delay search limits. N is the length of the frame. US 6,199,035 at col. 2, lines 17-42. Similarly, European Patent Application 0 628 947 A1, published Dec. 14, 1994, discloses techniques for coding of speech signal and specifically discusses the determination of a pitch period associated with a speech signal using a correlation-based long-term analysis:

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	Residual signal $r_s(n)$ is provided to a low-pass filter FPB, which generates a filtered residual signal $r_f(n)$ which is supplied to long-term analysis circuits LT1, LT2 estimating respectively pitch period d and long-term prediction coefficient b and gain G. Low-pass filtering makes these operations easier and more reliable, as a person skilled in the art knows. Pitch period (or long-term analysis delay) d has values ranging between a maximum d_H and a minimum d_L, e.g. 147 and 20. Circuit LT1 estimates period d on the basis of the covariance function of the filtered residual signal, said function being weighted, according to the invention, by means of a suitable window which will be later discussed. Period d is generally estimated by searching the maximum of the autocorrelation function of the filtered residual $r_f(n)$ $$R(d) = \sum_{n=0}^{Lf-1-d} r_f(n+d) \cdot r_f(n) \qquad (d = d_L \ldots d_H) \qquad (1)$$ EP 0 628 947 A1 at p. 3, lines 45-55. In another example, the background section of US Patent 5,553,191, filed Jan. 26, 1993 and issued Sept. 3, 1996, explains that estimating a pitch parameter of a speech signal using an open-loop, correlation-based, long-term prediction analysis has been known since at least as early as 1989: “It is previously known to determine a long term predictor, also called "pitch predictor" or adaptive code book in a so called closed loop analysis in a speech coder (W. Kleijn, D. Krasinski, R. Ketchum "Improved speech quality and efficient vector quantization in SELP", IEEE ICASSP-88, New York, 1988). This can for instance be done in a coder of CELP type (CELP=Code Excited Linear Predictive coder). In this type of analysis the actual speech signal vector is compared to an estimated vector formed by excitation of a synthesis filter with an excitation vector containing samples from previously determined excitation vectors. It is also previously known to determine the long term predictor in a so called open loop analysis (R. Ramachandran, P. Kabal "Pitch prediction filters in speech coding", IEEE Trans. ASSP Vol. 37, No. 4, April 1989), in which the speech signal vector that is to be coded is compared to delayed speech signal vectors for estimating periodic features of the speech signal.” US Patent 5,553,191 at col. 1, lines 16-33.

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Claims	Invalidity based on US 6,424,938
4. The extended signal codec of claim 1, wherein the at least one characteristic of the speech signal corresponds to a background noise level.	Defendants hereby incorporate by reference the disclosures above for claim 1. The '938 patent discloses the extended signal codec of claim 1, wherein the at least one characteristic of the speech signal corresponds to a background noise level. For example, and without limitation: For many common types of background noise a much lower bit rate than is needed for speech provides a good enough model of the signal. Existing mobile systems make use of this fact by downwardly adjusting the transmitted bit rate during background noise. For example, in conventional systems using continuous transmission techniques, a variable rate (VR) speech coder may use its lowest bit rate. '938 Patent at col 1, lns. 59-65. In FIG. 1, the input audio signal is input to a complex signal activity detector (CAD) and also to a voice activity detector (VAD). The complex signal activity detector CAD is responsive to the audio input signal to perform a relevancy analysis that determines whether the input signal includes information that is perceptually relevant to the listener, and provide a set of signal relevancy parameters to the VAD. The VAD uses these signal relevancy parameters in conjunction with the received audio input signal in order to determine whether the input audio signal is speech or noise. The VAD operates as a speech/noise classifier; and provides as an output a speech/noise indication. The CAD receives the speech/noise indication as an input. The CAD is responsive to the speech/noise indication and the input audio signal to produce a set of complex signal flags which are output to a hangover logic section which also receives as an input the speech/noise indication provided by the VAD. '938 Patent at col 3, line 55 – Col. 4, line 5. "The input signal (which can be preprocessed) is analyzed in the CAD by extracting information each frame about the correlation of the signal in a specific frequency band. This can be accomplished by first filtering the signal with a suitable filter, e.g., a bandpass filter or a high pass filter. This filter weighs the frequency bands which contain most of the energy of interest in the analysis. Typically, the low frequency region should be filtered out in order to de-emphasize the strong low frequency contents of, e.g., car noise. The filtered signal can then be passed to an open-loop long term prediction (LTP) correlation analysis. The LTP analysis provides as a result a vector of correlation values or normalized gain values; one value per correlation shift. The shift range may be, for example, [20, 147] as in conventional LTP analysis. An alternative, low complexity, method to achieve the desired relevancy

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	detection is to use the unfiltered signal in the correlation calculation and modify the correlation values by an algorithmically similar "filtering" process, as described in detail below. '938 Patent at col. 4, lines 19-37.
7. The extended signal codec of claim 1, wherein the at least one characteristic of the speech signal relates to backward linear prediction coding.	Claim 7 of the '841 Patent is anticipated by the prior art, including at least the '938 Patent. Defendants hereby incorporate by reference the disclosures above for claim 1. The '938 patent discloses the extended signal codec of claim 1, wherein the at least one characteristic of the speech signal relates to backward linear prediction coding. Furthermore, the complex signal activity detector (CAD) described in the '938 Patent determines that a music signal is relevant based on a characteristic of the speech signal that corresponds to a pitch information, using long-term prediction analysis. For example, without limitation: In conventional block-based speech coders the incoming speech signal is divided into blocks called frames. For common 4 kHz telephony bandwidth applications a typical framelength is 20 ms or 160 samples. The frames are further divided into subframes, typically of length 5 ms or 40 samples. '938 Patent at col. 1, lines 26-31. When comfort noise is generated in the decoder in a conventional DTX system, the noise is often perceived as being very static and much different from the background noise generated in active (non-DTX) mode. The reason for this perception is that DTX SID frames are not sent to the receiver as often as normal speech frames. In conventional linear prediction analysis-by-synthesis (LPAS) codecs having a DTX mode, the spectrum and energy of the background noise are typically estimated over several frames (for example, averaged), and the estimated parameters are then quantized and transmitted in SID frames over the channel to the decoder. '938 Patent at col. 2, lines 11-22. It can be seen from the foregoing that it is desirable to reduce the misclassification of subjectively relevant signals, while still maintaining a low bit rate (high compression) where appropriate, for example when compressing background noise while the speaker is silent. Very strong compression techniques can be used, provided they are not perceived as irritating. The use of comfort noise

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	parameters as described above with respect to DTX systems is an example of a strong compression technique, as is conventional low rate linear predictive coding (LPC) using random excitation methods. Coding techniques such as these, which utilize strong compression, can typically reproduce accurately only perceptually simple noise types such as stationary car noise, street noise, restaurant noise (babble) and other similar signals. ‘938 Patent at col. 2, lines 45-59. “According to the present invention, complex signal activity detection is provided for reliably detecting complex non-speech signals that include relevant information that is perceptually important to the listener. Examples of complex non-speech signals that can be reliably detected include music, music on-hold, speech and music combined, music in the background, and other tonal or harmonic sounds.” ‘938 Patent at col. 3, lines 7-13. “The input signal (which can be preprocessed) is analyzed in the CAD by extracting information each frame about the correlation of the signal in a specific frequency band. This can be accomplished by first filtering the signal with a suitable filter, e.g., a bandpass filter or a high pass filter. This filter weighs the frequency bands which contain most of the energy of interest in the analysis. Typically, the low frequency region should be filtered out in order to de-emphasize the strong low frequency contents of, e.g., car noise. The filtered signal can then be passed to an open-loop long term prediction (LTP) correlation analysis. The LTP analysis provides as a result a vector of correlation values or normalized gain values; one value per correlation shift. The shift range may be, for example, [20, 147] as in conventional LTP analysis. An alternative, low complexity, method to achieve the desired relevancy detection is to use the unfiltered signal in the correlation calculation and modify the correlation values by an algorithmically similar "filtering" process, as described in detail below. For each analysis frame, the normalized correlation value (gain value) having the largest magnitude is selected and buffered. The shift (corresponding to the LTP lag of the selected correlation value) is not used. The values are further analyzed to provide a vector of Signal Relevancy Parameters which is sent to the VAD for use by the background noise estimation process. The buffered correlation values are also processed and used to make a definitive decision as to whether the signal is relevant (i.e., has perceptual importance) and whether the VAD decision is reliable. A set of flags, VAD_fail_long and

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	VAD_fail_short, are produced to indicate when it is likely that the VAD will make a severe misclassification, that is, a noise classification when perceptually relevant information is in fact present." '938 Patent at col. col. 4, lines 19-52. It was well known to those of ordinary skill in the art of speech coding, well before the filing date of the '841 Patent, that long-term prediction (LTP) analysis (or pitch prediction) may be performed using either forward or backward linear prediction. Thus, the '938 Patent's teaching of the use of open-loop long term prediction (LTP) correlation analysis would be understood by one of ordinary skill in the art to inherently relate to backward linear prediction coding, as recited in claim 7. Examples of prior art teachings that evidence this common understanding of those skilled in the vocoder art include, without limitation:

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	In this paper we study the performance and the error sensitivities of six CELP [1] based codecs operating between 8 and 4 kbits/s. Codecs using both forward and backward adaption of the linear prediction coefficients and the long term predictor (LTP) are described. Initially we describe four low delay codecs which all use backward adaption of the LPC coefficients but which differ in their use of LTP. These codecs all have frame-lengths of 3 ms or less, and their performance at various bit rates between 8 and 4 kbits/s is examined. Next the error sensitivity of these codecs, and means of improving it, are described. Then an algebraic CELP (ACELP) [2] codec operating at 6.2 kbits/s with a frame-length of 5 ms is described. Our final codec also uses ACELP and operates between 4.7 and 7.1 kbits/s, but it is forward adaptive and so it has a much longer frame-length of up to 30 ms. After describing this codec we compare the performance of our codecs in both error-free conditions and in the presence of channel errors. Surprisingly the error sensitivity of the low delay backward adaptive codec with no LTP is similar to that of the forward adaptive, high delay, ACELP codec. © 1997 Academic Press Woodward, J.P., and Hanzo, L., A Range of Low and High Delay CELP Speech Codecs Between 8 and 4 kbits/s, *Digital Signal Processing* **7** (1997), pp. 37-46, at Abstract. “A backward-adaptive pitch prediction algorithm is described. It is used in conjunction with a backward-adaptive short-term predictor in a low-delay speech coding system operating at 16 kb/s. The backward-adaptive pitch prediction algorithm is a hybrid algorithm which combines backward block adaptive pitch prediction and backward recursive pitch prediction. The pitch predictor tap gains and the pitch period are periodically initialized by using a backward block adaptive algorithm. Between these initializations, however, both the tap gains and the pitch period are adapted using backward recursive algorithms. The tap gains are adapted using the well-known gradient algorithm, in a manner similar to the way the short-

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	term predictor coefficients are adapted. The pitch period is adapted using a novel pitch tracking algorithm. By combining backward recursive adaptation with backward block adaptation, it was possible to increase the prediction gain of the pitch predictor and reduce the interval required between initialization of the pitch predictor parameters." Pettigrew, R.; Cuperman, V., "Backward pitch prediction for low-delay speech coding," *Global Telecommunications Conference, 1989, and Exhibition. Communications Technology for the 1990s and Beyond. GLOBECOM '89., IEEE* , vol. 2, pp.1247-1252, 27-30 Nov 1989.